ABSTRACT

A system for processing audio signals comprises a sequence of digital filters each configured to process a selected frequency using a set of coefficients. A filter configured to process a certain frequency shares its coefficients with another filter that processes a frequency that is lower than the first frequency by at least one frequency interval, such as an octave. The first filter samples at a certain sampling rate, and the second filter's sampling rate is determined by multiplying the first sampling rate by the ratio of the second frequency to the first frequency. The filters are evenly grouped into frequency intervals, such as octaves. Filters in an octave are sampled at a sampling frequency that is at least twice as high as the highest frequency processed in that octave.